

# DIGITAL TELEVISION TRANSMISSION AND BROADBAND NETWORK TECHNOLOGIES

*Sonja Grgić, Mislav Grgić, Branka Zovko-Cihlar*

Department of Radiocommunications and Microwave Engineering  
Faculty of Electrical Engineering and Computing, University of Zagreb  
Address: FER, Unska 3, HR-10000 Zagreb, Croatia  
Phone: + 385 1 6129 780, Fax: + 385 1 6129 717

**Abstract** - *Digital television (DTV) network is medium that transports moving images, in the form of digital video signal and accompanying sound, in the form of digital audio signal, from one point to another. However, the system may include data and control signals and hence may be thought of as a multimedia system. Various network technologies find use in the digital television production and transmission of digital television signals. All transmission media are analog channels, but binary digital signals are so tolerant of channel characteristics that they are easily handled. The capability of most communication channels is much greater than what binary transmission requires. Many methods are available for digital television transmission over various channel types for different applications and picture quality requirements. These are the subjects of this paper. The important topics such as: network properties, quality of picture requirements, video interface design, aspects of coding and new transmission formats, will be taken into consideration.*

**Keywords:** Television Networks, Digital Television, Contribution, Distribution, Digital Video Broadcasting

## 1 INTRODUCTION

Television networks carrying television signals need to transport video and accompanying audio signals. Historically, analog television networks have played an important role. Transmission characteristics and impairments (noise, interference, distortion, instability) in these networks are well known. Digital television technology performs quite differently. The impairments are not related to distance as with analog transmission but depend very much on codec technology and interface specifications. These networks allow better and easier TV production and high-quality TV transmission.

It is possible to characterise the purposes for which digital television network will be used. The DTV networks can be classified as:

- (a) networks for contribution purposes
- (b) networks for distribution purposes.

A contribution network is a network at the end of which there may be a need for high-quality picture processing (e.g. inter studio links). This means that the transmission system should not (ideally) limit the processing possibilities that can be performed on the signal. A distribution network is a network at the end of which there is likely to be little or no need for high-quality picture processing (e.g. studio-transmitter links). This means that the emphasis in the design of the system is on the highest subjective picture quality.

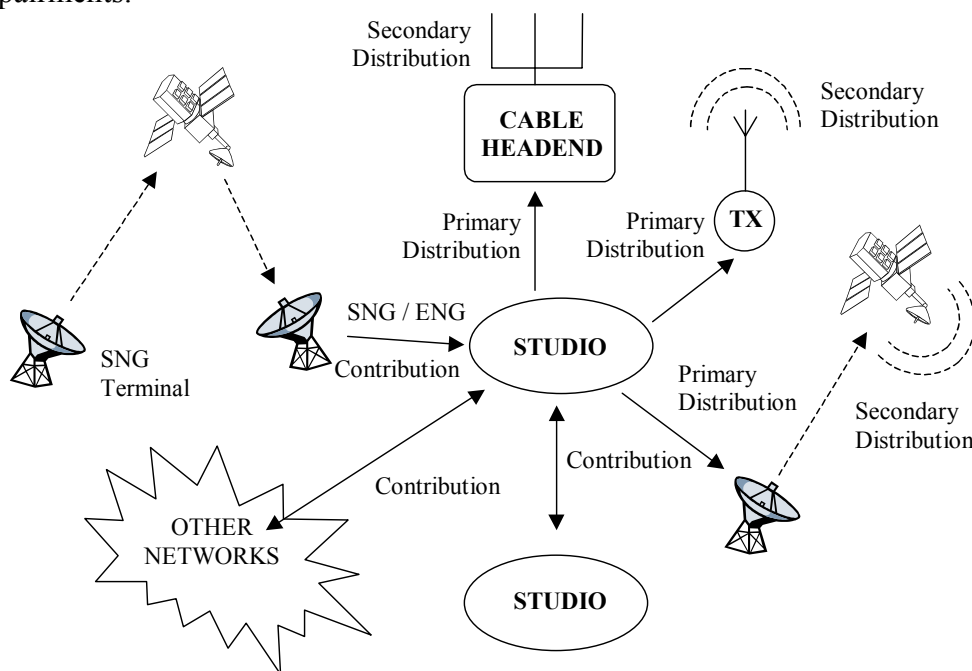
The architecture of a DTV network is shown in Figure 1. TV contribution include various elements:

- studio internal distribution - high quality transfer of video and audio materials suited for further processing;
- contribution between studios;
- contribution from the field - electronic and satellite news gathering (ENG/SNG).

In distribution there are the following two areas:

- primary distribution: the links that feed signals to the transmission sites (terrestrial transmitters, cable head-ends and satellite up-links) from the studio, probably via a switching centre;
- secondary distribution: the links that radiate from the cable TV head-end or path from a satellite uplink and beyond.

The network technologies that are optimal for contribution of television signals are not the same as those optimal for distribution. In the studio environment, a transparent video transmission is needed. A process or a link is considered transparent if it does not deteriorate the source material in either the perceived image quality or its suitability for further processing. The first stages of TV production require processing and transmission that will not significantly deteriorate the source material [1]. In that respect digital television technology holds a highly important position as it ensures predictable performance of the video processing and allows transmission and storage without impairments.

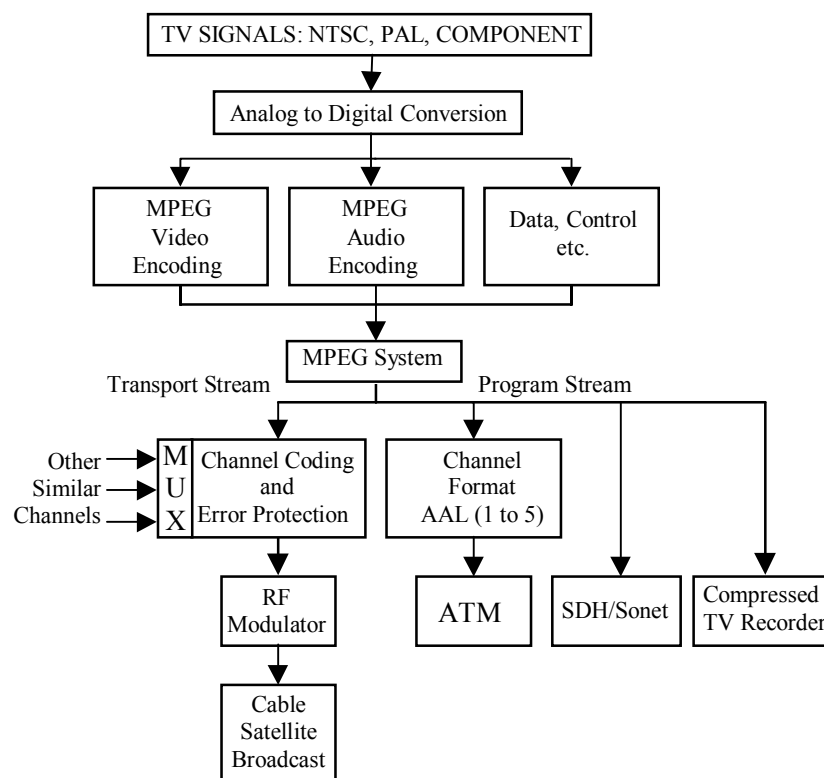


**Figure 1** Television networks for contribution and distribution purposes

## 2 STRUCTURE OF DIGITAL TELEVISION NETWORK

Television is rendered in digital form for the inherent quality and lack of degradation in replication and transmission. The digital TV signal is usually compressed for efficiency in transmission and storage [2]. The consequence of signal compression is bit-rate reduction (BRR). The block-diagram of compressed television processing and transmission is shown in Figure 2. Video inputs are analog NTSC, PAL or component. Video and audio are converted to digital domain in accordance with a sampling standard such as 4:2:2 video source standard defined in ITU-R Rec.601 [3]. The data rate of a 4:2:2 signal (10-bit word length) is 270 Mb/s. This full-bandwidth digital video is an extremely important part of the television system. Program production processes should be full-bandwidth digital in order to manipulate the video to produce desired results. Formatting and studio interconnection of digital television signal follows a related standard ITU-R Rec.656 for serial digital interconnects (SDI) [4]. It is worldwide standard used for serial digital video transmission. ITU-R Rec.656 was defined for component signals and supported 8-bit word length. Therefore, the alternative approach has been developed to allow up to 10-bit samples of component video to be communicated serially. The proposal is now well established as the SDI and published as standard SMPTE 259M in August 1995.

Following program production, television signal may be compressed for storage, efficient transmission and studio internal distribution. Typically it will be MPEG-2 compression [5], resulting in an MPEG transport stream (MTS) that may be multiplexed with other MPEG transport streams for transmission and interconnection. The MPEG-2 is a standard that defines *generic syntax* for transport of video and associated audio by means of data compression and supports a wide range of performance levels and parameters (resolution, bit-rate, frame rate, etc.). MPEG-2 coding scheme covers video coding, audio coding and system data formatting. Digital data from each of the coders and other sources is formed into either a transport or a program stream. The transport stream uses fixed-length packets, while the program stream uses variable-length packets. A transport stream may carry information from more than one program while the program stream has data associated with only one program. The transport stream with its fixed-length packets is specially designed for longer-range transmission. The program stream, which is designed for use in relatively error-free environments, can be used as input to a digital recorder and to various types of telecommunication channels. Because of the characteristics of media-based recording, special formatting, channel coding and error protection are built into TV recorders. This makes the record/playback look like an error-free process. Traditional wire or fibre communication channels have relatively low error rates. However, some optional error protection may be required to meet the error-free criteria. Not all of the transmission possibilities are shown in Figure 2. The transport stream could be used where the program stream is used. Also, there can be multiplexing of data streams for both recorders and telecommunications channels.



**Figure 2** Compressed television processing and transmission

A number of telecommunications transport methods can be used for compressed television. The synchronous digital hierarchy (SDH) is a set of telephone standards enabling synchronous multiplexing of data streams on high-speed links. SDH can directly carry MPEG-2 transport stream with simple data formatting, although there is presently no standard. Looking toward the future, Asynchronous Transfer Mode (ATM) is a likely candidate for transmission of packetized data certainly for long distances and perhaps for contribution purposes within the studio [6]. MPEG-2 is designed to be compatible with the ATM transport mechanisms. To utilise ATM, a special ATM

adaptation layer (AAL 1 to 5) is used to map the 188-byte MPEG transport stream packets into 53-byte ATM packets. Digital video broadcasting (DVB) systems include radio-frequency (RF), cable and satellite transmission methods to carry the MPEG transport stream data [7]. Every type of channel has its own channel coding methods and may include error protection. Multiplexing of a number of data streams on to one transmission channel is one of the most common uses of compressed TV today. The characteristics of the channel coder are selected to support the modulation scheme and the transmission medium through which the data must be transported.

One of the advantages of MPEG-2 compression is the capability of providing different video picture quality levels based on bit rates. Distribution quality to the home may be adequate with bit rates of 2 to 8 Mb/s for standard definition television (SDTV) and 20 Mb/s for high definition television (HDTV). Contribution quality is required if further production processing is planned for the decompressed video. In that case, bit rates of 18 to 50 Mb/s are required for SDTV and 200 to 400 Mb/s for HDTV.

The Open System Interconnection (OSI) model was developed by International Standard Organisation (ISO) as standard IS 7498 (in 1984) for communication systems and provides a layered analysis tool that can be applied to various related technologies. A mapping of ISO OSI 7-layer standard on television transmission is shown in Table 1. It is a useful technique for identifying the interface specifications necessary to provide independence between different aspects of a communications system. Emphasis is placed on MPEG-2 compressed data (see Figure 2) that is becoming the predominant method for video.

**Table 1** ISO OSI seven layer reference model

Layer	Traditional definition	Television transmission	Implementation example
Application	User determined	Programming, effects, recording, contribution, distribution	Play list
Presentation	Data transformation	Signal coding, compression, file systems	Encryption, MPEG-2 compression
Session	Connection management	Control, access management	MPEG systems
Transport	End-to-end management	Mux./demux., quality of service, synchronisation	MPEG-2 transport, Rec.656
Network	Subnet management	Routing, switching, network protocols	Switching
Data link	Error management and synchronisation	Error detection/correction, acknowledgement, framing	SDI synchronisation
Physical	Mechanical, electrical, functional	Signal levels, impedances, connectors, jitters	Twisted pair, coaxial cable, optical fibre, radio

### 3 NETWORKS FOR CONTRIBUTION PURPOSES

#### 3.1 Technical criteria for studio operations

The fundamental technical criteria for successful studio operations include bit rate (throughput) and storage size, real-time signal flow, deterministic control, signal integrity and security and criteria specific to compression.

The *bit rate* requirements of video undersize those of any other data type. A rate of about 20 Mb/s and upwards is of interest for professional production work. When it is not compressed the well-known rate of 270 Mb/s applies for SDTV. Every process, buffer or silicon chip that the signal passes through must be able to sustain these rates. It also means that *storage capacities* for full-length films extend to hundreds of gigabyte [8].

Television signals today are handled in both real-time and non-real-time processes. When a signal is transmitted or presented in *real-time* the average delay between destination and sending end must be constant and variation in delay (jitters) must be low enough that no television frames are skipped, frozen, lost or duplicated. Where the transmission is *non-real-time* it must still respect the synchronous nature of source information and allow the original information to be properly reconstructed [9].

The production of television requires that a director should be able to stop presentation of one flow of visual information and switch to another. It can be performed live as an edit. A live switch must occur with no apparent delay from the issuance of the command to its execution. In an edit a delay may be accepted but it must be constant. These requirements of low latency and constant latency define *deterministic control*. Not all networks can make deterministic switch.

**Table 2** Categories of digital video compression

Category	Bit Rate	Main applications
Linear	140/155 Mb/s	Contribution, production
Medium degree of BRR	34/45 Mb/s	Contribution, primary distribution
High degree of BRR	2-8 Mb/s MPEG-2	SNG, secondary distribution

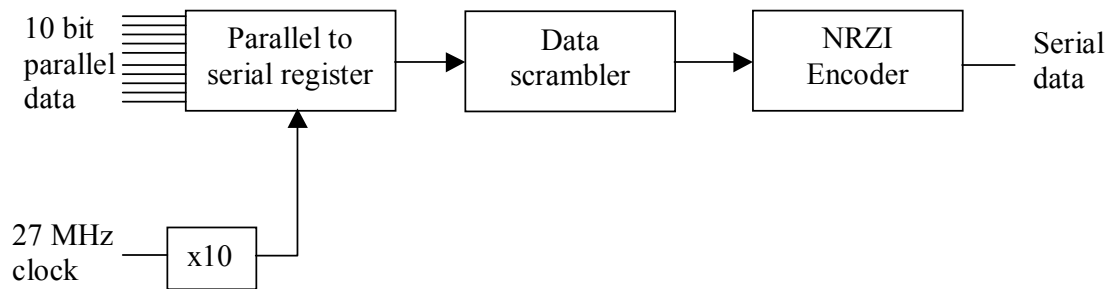
*Digital compression criteria* must address the topics of digital TV production (such as lossless duplication and editing and low-loss processing of compressed video) and the emerging network services. It may be necessary or desirable to use one category of compression algorithms in one video generation followed by different category member in the next and subsequent generation. The categories of compression algorithms are in Table 2. It is essential that overall losses are not exaggerated by a mismatch the two different compression variants. The available network interfaces must reflect categories shown in Table 2.

### 3.2 Network technologies

#### 3.2.1 Parallel and serial digital connections

For the highest-quality production and post-production studio operation, digital interconnection between such items as tape or optical players/recorders, disc recorders, video/audio servers, codecs, video/audio processors and workstations is highly desirable to avoid degradation due to repeated conversion and filtering stages. There are two forms of digital video interconnection standards: parallel and serial. The parallel connection uses a 25-wire multiconductor cable and 25-pin connectors. One pair of wires is required to carry the information for a single bit. The clock is also carries in a pair of wires. In 4:2:2 sampling structure according to ITU-R Rec.601 the luminance is sampled at 13.5 MHz and two color difference signals are sampled at one-half that frequency. All of the signals use 8 or 10 bit resolution. The entire digital video signal can be carried as a 27 MHz parallel data stream. For short distances this approach represents the optimum solution. Distances of up to 50m are allowed. Beyond that point, the data and related clock information begin to "skew" in time because of their differing frequency content. The clock and data information arrives at different times, causing data errors. In addition cable and connectors are physically large, difficult to use in a large facility and expensive to implement. The answer for these problems is the serial

connection. All of the digital samples are shift registered using a 10 times parallel rate clock, Figure 3. The resulting 1 bit wide serial data is passed through a data scrambler and coded to form a self-clocking channel code. A known algorithm is applied to ensure that there is sufficient number of transitions to recover the clock signal at the receiving end and to limit any DC content on the signal. The data is then passed through a nonreturn to zero inverted (NRZI) encoder, which takes a transition in the signal and changes it to a 0 and takes a nontransition in the signal and changes it to a 1. This further limits any DC content in the signal. The serial data is then sent down the coaxial cable. The electronic complexity of a serial interconnect is greater than complexity of parallel interconnects. The data rate for this distribution method is given as number of bits per sample times parallel transmission line rate. When we serialise the 27 MHz signal in 10-bit word length application, the resulting serial data rate is 270 Mb/s.



**Figure 3** Serial transmitter

The primary advantage of serial digital approach is simplified interconnections. Coaxial cable and fibre optics can be used as physical media. Serial connection may use existing analog cables but suitable equalisation and termination must be provided. The transmission rates for serial digital transmission are in the RF spectrum. Termination impedance will require a high degree of accuracy to minimise reflections on the cable. SDI offers 270 Mb/s clock rate using up to 200 m of coaxial cable or up to 20 km on single mode optical fibre.

An advantage of serial transmission is that a matrix distribution unit is more easily realised. Where numerous pieces of video equipment need to be interconnected in various ways for different purposes, a crosspoint matrix is an obvious solution. With serial signals, only one switching element per signal is needed, whereas in parallel system, a matrix would be unwieldy. SDI has the ability to perform high speed routing via serial digital switching matrices. A typical SDI routing matrix of size 64×64 offers a switching bandwidth of 17.28 Gb/s.

Although the SDI is accepted method of transmitting baseband serial digital video between ITU-R Rec.601 compliant equipment the issue of compressed video has brought a desire to transmit the compressed video signals in the bit-stream format. One possible solution is Serial Digital Data Interface (SDDI). It is intended to be used to transport packetised data within a studio/production centre environment. SDDI can transmit multiple compressed video signals on one SDI link. SDDI has the advantages of being compatible with well-established SDI (SMPTE 259M) standard. SDDI is being proposed to the SMPTE as a standard for broadcast use.

### **3.2.2 Telecommunication and computer networks**

The world of telecommunications and computers has numerous methods of high-speed serial networking. The Plesiochronous Digital Hierarchy (PDH) is a specification of a set of digital transmission streams that are organised hierarchically from one 64 kb/s telephony channel (Table 3). The use of PDH as a medium for video transmission has some disadvantages. It is necessary to perform complete demultiplexing and multiplexing on the whole hierarchy in order to pick out or insert a data stream. The structure of multiplexers makes difficult to measure network performance and to manage remote network elements.

The SDH (also known as SONET in USA) offers a series of important improvements. It constitutes a transparent multipurpose transport network on which various services may be based. SDH operates at a number of fixed data rates: 51.84 Mb/s (called the Synchronous Transfer Module STS-1), 155.25 Mb/s (STS-3) and 622.08 Mb/s (STS-12). Thanks to the management functions, the individual routes can be supervised and paths may be redirected if necessary. SDH provides a backbone structure onto which other systems can be mounted. Of these, the most important is ATM [12]. Another possible system is Fibre-Channel that uses the SDH hierarchy but has a different packet structure compared to ATM.

**Table 3** European and American Plesiochronous Digital Hierarchy

Europe	USA
DS0-64 kb/s	DS0-64 kb/s
30xDS0→E-1 - 2.048 Mb/s	24xDS0→DS1 or T-1 - 1.544 Mb/s
4xE-1→E-2 - 8.448 Mb/s	4xT-1→DS2 or T-2- 6.312 Mb/s
4xE-2→E-3 - 34.368 Mb/s	7xT-2→DS3 or T-3- 44.376 Mb/s
4xE-3→E-4 - 139.264 Mb/s	6xT-3→274.176 Mb/s*
4xE-4→E-5 - 565.148 Mb/s*	

\*Rates not specified by ITU

ATM is a protocol, which puts payload's packet into 53-byte cells and has end-to-end addressing (virtual paths and virtual circuits). The protocol is characterised by switched-circuit technology and variable bandwidth. The fixed cells accommodate mixing of various applications, allow prediction of network delays and enable traffic from multiple sources to be switched to multiple destinations. ATM is only a protocol, not a transmission system - it needs a physical layer to ride on. An ATM interface would typically be an SDH interface or perhaps an E-3/T-3 interface. Some ATM interface rates are shown in Table 4.

**Table 4** ATM interface rates

Type	Bit rate (Mb/s)	Physical medium
SONET 12	622.08	Optical
SONET STS-3	155.52	Optical/coax
SONET STS-1	51.84	UTP
Cell based	155.52	Optical/STS/UTP
Cell based	100.00	Optical

UTP=unshielded twisted pair

STP=shielded twisted pair

Starting with the 270 Mb/s uncompressed sources, MPEG-2 first compresses them, then uses its own defined transport layer to form a Constant bit Rate (CBR) Transport Stream consisting of 188-byte packets from the incoming material. ATM Adaptation Layer (AAL) provides the conversion between MPEG's 188-byte packets and 53-octet cell format [13]. Different versions of this layer exists for the various kinds of ATM traffic types; the one standardised for CBR MPEG-2 video is known as AAL-5. The ATM cells can be packed into an SDH container. This is just one way of transporting ATM cells. An alternative interface is "Cell Based" in which the cells are more directly mapped onto the physical layer. The advantage of the SDH interface is that the cells are ready to leave the Local Area Network (LAN) and go onto Wide Area Network (WAN). There is now effectively no limit to where these signals can go. Instead of serving just one or more production suites in the same building, city-wide or country-wide facilities can be interconnected.

The transmission of component-coded digital television signals for contribution quality applications at bit-rates near 140 Mb/s is defined by CCIR Rec.721 [10]. The CCIR Rec.721 coding scheme is intended for contribution of CCIR Rec. 601 interlaced TV signals aiming at very high quality (almost lossless quality). The CCIR Rec.723 [11] specifies digital coding of component television signals for contribution quality applications at bit-rates in range 34-45 Mb/s. The image quality is sufficient for limited post-production purposes (see Table 4).

The computer world offers many networking technologies [14]. Most have been too slow to be useful for video data rates. However, the latest standards offer at least 100 Mb/s basic data rate. The examples are: Firewire (also known as P1394 or the Home Video Bus) operating at 100 Mb/s and 400 Mb/s, Fibre Distributed Digital Interface (FDDI) operating at 100 Mb/s and Ethernet operated originally at 10 Mb/s, now available at 100 Mb/s. Most of these networks fail to offer easy access to real-time networking.

### 3.2.3 Video data file transfer and streaming

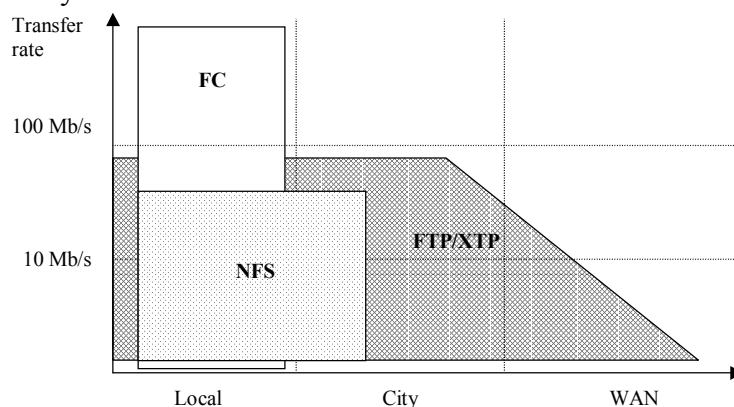
File transfer and streaming (content play) are the most common functions needed by users of digital content within a production facility. This implies a local area network environment. Wide area networks offer the possibility of incorporating remote storage devices within a logical cluster (distributed video production). Although technology from the networking and computer sectors is useful for moving digital files, moving digital video/audio program material in a frame synchronised, fixed delay, real-time way with little or no loss is a foreign concept to most users of networks and computers.

The file transfer is the transfer of any data wrapped with a known cover. The wrappers are used to group and label the program material and related information. Program material and related information is called content. Wrappers are intended to be used to link physical media together for streaming of content across interconnects and to store content in file systems and on servers. The content play is raw data that is streamed from a serving device to one or more receivers. Typically this raw data represents audio and video.

The two main methods of sending data are:

- guaranteed-delivery "File Transfer" methods
- bounded-quality, asset streaming ("Content Play") transfer methods. Digital content may be streamed from a serving device at user-selectable payload rates for real-time or slower/faster than real-time.

Guaranteed delivery indicates that the entire payload will reach the destination without bit-errors, barring and failure of the physical link. The term "Bounded quality" is used for a transfer method that is designed to move the payload from source to destination(s) but without the absolute certainty of true guaranteed delivery.



**Figure 4** File transfer applications

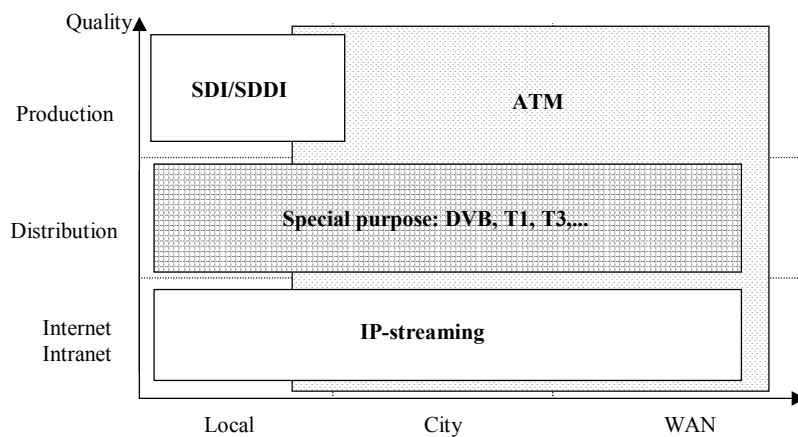


A pure file transfer needs to meet the following minimum user requirements: error-free transfer, the connection between sender and receiver must be bi-directional, file transfer "as fast as possible (AFAP)" must be supported, slow and fast transfer speeds are required, files access and permission control are required, LAN and WAN should be supported that implies a routable network layer like Internet Protocol, point-to-point and point-to-multipoint transfers must be supported. There is no single proven file transfer method available today to perform all of the user requirements. Some possible technologies are: file transfer protocol (FTP), express transport protocol (XTP), fibre channel (FC) and network file system (NFS). Figure 4 maps various application spaces to file transfer/access technology.

A video streaming has to meet the special user requirement of playing video, audio, data etc. in a television production environment. Because re-sends of video data (and resulting delay and latency) caused by corrupted data cannot be tolerated, some errors must be accepted and a return path must not be used. Other important unique television production requirements are: bounded quality based on the quality of service of the link, the AFAP mode is not required, streaming rates may be at real time or faster or slower than real time with real time being default mode, links must be unidirectional in nature, file access and permission control are required, receivers may "join" the stream at any time without needing to start at the beginning, point to multipoint transfers are common. Video streaming in production and distribution environments can be split into two main categories

- general purpose – IP streaming, ATM over a variety of physical layers, SDI/SDDI;
- special purpose - IEEE 1394, T1, T3, E1, E3, DVB, FC.

Figure 5 maps various application spaces to streaming technology. SDI is usually chosen as the primary way to stream video content. ATM can be used to connect production facilities but find less use inside the studio. IP-based audio/video streaming is not mature enough to be selected as a proffered way.

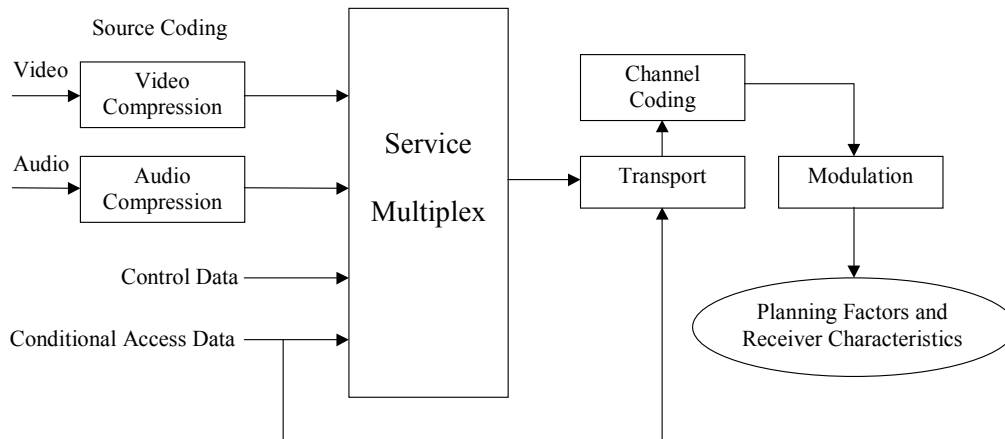


**Figure 5** Streaming technology applications

## 4 DIGITAL TV BROADCASTING

Television broadcasting systems using digital modulation techniques provide several benefits over systems employing analog techniques. Digital television systems can provide more immunity from interference and, therefore, can provide a more secure communications link than analog systems. Digital communications systems can be more spectrum efficient than analog systems, as they are capable of transmitting more information per channel. In digital communications systems the information is in form that facilitates encoding and encryption for applications that require conditional access and security. The advent of digital television systems transforms the classical TV channel to a data transmission medium which can carry huge data rates at extremely low bit error rates (below  $1 \cdot 10^{-11}$ ).

Over the past five years there has been an intensive activity in Europe to finalise the standards for digital TV broadcasting by cable, satellite and terrestrial networks. The technical specifications are elaborated by the Digital Video Broadcasting (DVB) Project conducted under the auspices of the European Broadcasting Union (EBU). They are next passed to the European Telecommunications Standards Institute (ETSI) for further procedures toward publication of the standards. The specifications were finalised in December 1993 for digital satellite broadcasting (DVB-S) [15] and in January 1994 for digital cable services (DVB-C) [16]. The most complex of all DVB transmission systems is the one to be used in terrestrial channels (DVB-T). This specification was finalised in early 1997 [17].



**Figure 6** DVB system model

The common system model for different DVB specifications (DVB-S, DVB-C and DVB-T) is shown in Figure 6. The system model can be divided into four subsystems: source coding and compression, service multiplex and transport, the physical layer (modulation scheme) including the channel coding parameters, planning factors and implementation strategies. Source coding refers to coding methods to reduce the very large data stream created when images are represented by digital video samples or when sound is represented by digital audio samples. The service multiplex and transport refers to the way of dividing the digital data stream into packets of information and the appropriate methods of multiplexing the video data stream, the audio data stream packets and control data steam packets into a single data stream to construct a single program. Multiplexing also provides the capability of combining different program data streams into a single broadcast channel for simultaneous delivery. In developing transport mechanism interoperability between digital media such as terrestrial broadcasting, cable distribution, satellite distribution, recording media and computer interfaces must be prime consideration. The DVB project decided that for source coding of audio and video signals as well as multiplexing of the different signals for digital broadcasting systems (DVB-C, DVB-S and DVB-T) the MPEG-2 standard should be used. A channel coder takes the resulting compressed data bit streams and adds additional information that can be used by receiver to recognise and reconstruct the images, sound and control data from the transmitted signal. This module includes mechanisms by which additional data are added to the multiplexed data stream to provide protection against loss of the signal. The characteristics of channel coder are selected to support the modulation scheme and the medium through which the data must be transported. Modulation is a mechanism whereby the protected data stream is imposed on one or more carrier signals for transmission. Planning factors and implementation strategies include consideration of the characteristics of the transmission media and receivers.

The design of transmission standards for the different broadcast transmission media requires in-depth understanding of the channel characteristics of the different media. Whereas the satellite channel, for instance, is a wide-band channel (33 MHz of bandwidth or so) and very much

power-limited as well as strongly influenced by the nonlinearities of the power amplifier onboard the satellite, the coaxial cable channel is narrowband (6-8 MHz) and power-limited only in the sense that the cumulative power of all signals on the cable must not exceed limitations set by the amplifiers in the cable network.

The standard for DVB-S applications is based on Quaternary Phase Shift Keying (QPSK) modulation. The standards are used for primary and secondary distributions in fixed satellite service (FSS) and broadcast satellite services (BSS) in the 14GHz/12GHz band. The standard defines all transmit and receive functions and characteristics except the symbol rate. The reason for this is that there are a variety of satellites with different transponder bandwidths.

The basic modulation scheme for DVB-C is Quadrature Amplitude Modulation (QAM) with 64, 32 or 16 states. Although the cable channel is 8 MHz in all European countries, no symbol rate was fixed for cable TV either. The reason for this is the desire to be able to feed satellite signals to cable networks without any processing at the cable head-end.

The standard for DVB-T employs Coded Orthogonal Frequency-Division Multiplex (COFDM) modulation technique. The major benefit of COFDM is that the serial baseband bit-stream, which needs to be transmitted, is distributed over many closely spaced individual carriers (frequency division). This spreading makes the signal robust against the effects of multipath and narrowband interferes. DVB-T is primarily tailored for 8 MHz channels but adaptation to 7 MHz or 6 MHz channels are possible. Looking at the DVB's key goal to develop a family of closely related systems for digital television, DVB-T is developed in such a way that the performance was as similar to that of DVB-S and DVB-C as possible to enable the manufacture of multistandard receivers at lowest possible cost.

## **5 CONCLUSION**

In more than 50 years of commercial broadcast television, few changes have affected the medium more than the conversion from analog standards to digital. Only now this change has been introduced in such a manner that entirely digital broadcast system can receive, process and deliver digital television signal to an end user. This opens the market for various types of broadcast products to be delivered from video-on-demand to interactive video data services. In addition, digital television network technology is undergoing drastic changes. Digital television networks are becoming data carriers that may support a multitude of applications. Digital networks are increasingly changing technology in order to support new services efficiently.

At the same time television broadcasting is facing a quest for more channels and new diversified services. New enabling technologies such as digital networking and bit-rate reduction techniques provide substantial momentum. Digital technologies are making contribution, distribution, storage and delivery of television signals easier but also creating new challenges for the designer of a digital facility. They have to consider some important topics:

- digital broadcast site need a long-term plan with standards that guarantee compatibility and an evolutionary path;
- in dealing with new digital medium challenges in compression, insertion, editing, storage and distribution must be addressed;
- there has been a shift in the digital broadcasting environment so that traditional broadcaster must become a data network specialist.

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